IN THE CLAIMS:

The text of all pending claims, (including withdrawn claims) is set forth below. Cancelled and not entered claims are indicated with claim number and status only. The claims as listed below show added text with underlining and deleted text with ~~strikethrough~~. The status of each claim is indicated with one of (original), (currently amended), (cancelled), (withdrawn), (new), (previously presented), or (not entered).

In accordance with the following, claims 1, 2, 3, 8, 10 and 11 have been amended. No claim has been cancelled or added.

1. (currently amended) An adaptive beamforming method, comprising:

compensating for time delays of M noise-containing speech signals input via a microphone array having M microphones, wherein M is an integer greater than or equal to 2, and generating a sum signal of the M compensated noise-containing speech signals; and

extracting pure noise components from the M compensated noise-containing speech signals using feedback providing a noise-removed signal to M adaptive blocking filters ~~that are connected to~~and M adaptive canceling filters connected in a feedback structure and ~~extracting pure speech components from the sum signal using the M adaptive canceling filters that are connected to the M adaptive blocking filters in the feedback structure~~finally generating the noise-removed signal from the sum signal by providing the pure noise components to the M adaptive canceling filters.

2. (currently amended) The method of claim 1, wherein the extracting of the pure noise components comprises:

filtering ~~a~~the noise-removed sum signal through the M adaptive blocking filters;

subtracting signals output from the M adaptive blocking filters from the M compensated noise-containing speech signals to output M noise signals;

filtering the M noise signals through the M adaptive canceling filters;

subtracting signals output from the M adaptive canceling filters from the sum signal and inputting M subtraction results to the M adaptive blocking filters as the noise-removed sum signal; and

adding the M subtraction results.

3. (currently amended) The method of claim 1, wherein the extracting of the pure noise ~~signals~~components comprises:

filtering ~~a~~the noise-removed sum signal through the M adaptive blocking filters;

subtracting signals output from the M adaptive blocking filters from the M compensated noise-containing speech signals to output M noise signals;

filtering the M noise signals through the M adaptive canceling filters;

adding signals output from the M adaptive canceling filters and outputting an adaptive canceling filter sum signal; and

subtracting the adaptive canceling filter sum signal from the sum signal and inputting M subtraction results to the M adaptive blocking filters as the noise-removed sum signal.

4. (original) The method of claim 2, wherein the M adaptive blocking filters and the M adaptive canceling filters are finite impulse response filters.

5. (original) The method of claim 4, wherein coefficients of the M adaptive blocking filters and the M adaptive canceling filters are updated by an information maximization algorithm.

6. (original) The method of claim 3, wherein the M adaptive blocking filters and the M adaptive canceling filters are finite impulse response filters.

7. (original) The method of claim 6, wherein coefficients of the M adaptive blocking filters and the M adaptive canceling filters are updated by an information maximization algorithm.

8. (currently amended) An adaptive beamforming apparatus, comprising:

a fixed beamformer that compensates for time delays of M noise-containing speech signals input via a microphone array having M microphones, wherein M is an integer greater than or equal to 2, and generates a sum signal of the M compensated noise-containing speech signals; and

a multi-channel signal separator that extracts pure noise components from the M compensated noise-containing speech signals using feedback providing a noise-removed signal to M adaptive blocking filters ~~that are connected to~~and M adaptive canceling filters connected in a feedback structure and ~~extracts pure speech components from the sum signal using the M adaptive canceling filters that are connected to the M adaptive blocking filters in the feedback structure~~finally generates the noise-removed signal from the sum signal by providing the pure noise components to the M adaptive canceling filters.

9. (original) The apparatus of claim 8, wherein the fixed beamformer comprises:

a time delay estimator that calculates time delays of the M noise-containing speech signals input via the microphone array;

a delay unit that delays the M noise-containing speech signals by the time delays calculated by the time delay estimator; and

a first adder that adds the M noise-containing speech signals delayed by the delay.

10. (currently amended) The apparatus of claim 8, wherein the multi-channel signal separator comprises:

a first filter that filters ~~a~~the noise-removed sum signal through the M adaptive blocking filters;

a first subtractor that subtracts signals output from the M adaptive blocking filters from the M compensated noise-containing speech signals using M subtractors;

a second filter that filters M subtraction results of the first subtractor through the M adaptive canceling filters;

a second subtractor that subtracts signals output from the M adaptive canceling filters from the sum signal using M subtractors, and inputs M subtraction results to the M adaptive blocking filters as the noise-removed sum signal; and

a second adder that adds signals output from the M subtractors of the second subtractor.

11. (currently amended) The apparatus of claim 8, wherein the multi-channel signal separator comprises:

a first filter that filters ~~a~~the noise-removed sum signal through the M adaptive blocking filters;

a first subtractor that subtracts signals output from the M adaptive blocking filters from the M compensated noise-containing speech signals using M subtractors;

a second filter that filters signals output from the M subtractors of the first subtractor through the M adaptive canceling filters;

a second adder that adds signals output from M adaptive canceling filters of the second filter; and

a second subtractor that subtracts signals output from the second adder from the signals output from the fixed beamformer and inputs M subtraction results to the M adaptive blocking filters as the noise-removed sum signal.

12. (original) The apparatus of claim 10, wherein the M adaptive blocking filters and the M adaptive canceling filters are finite impulse response filters.

13. (original) The apparatus of claim 12, wherein coefficients of the M adaptive blocking filters and the M adaptive canceling filters are updated by an information maximization algorithm.

14. (original) The apparatus of claim 11, wherein the M adaptive blocking filters and the M adaptive canceling filters are finite impulse response filters.

15. (original) The apparatus of claim 14, wherein coefficients of the M adaptive blocking filters and the M adaptive canceling filters are updated by an information maximization algorithm.

16. (original) An adaptive beamforming apparatus, comprising:

a receiver that receives signals including noise components, delays the received signals by a calculated time to provide delayed received signals, and adds the delayed received signals to provide a combination received signal;

a signal separator that generates a clean signal without noise components based on adaptively filtering the delayed received signals and the combination received signal by a plurality of adaptive blocking filters having blocking coefficients and a plurality of adaptive canceling filters having canceling coefficients connected in a feedback structure, wherein the blocking coefficients and the canceling coefficients are automatically updated during operation of the signal separator.

17. (original) The apparatus of claim 16, wherein the feedback structure of the signal separator comprises:

a plurality of first subtractors that receive the delayed received signals and subtract corresponding signals from the plurality of adaptive blocking filters to output separate noise component signals; and

a plurality of second subtractors that receive the combination received signal and subtract corresponding signals from the plurality of adaptive canceling filters to output separate clean signals without noise components, wherein the plurality of adaptive blocking filters receive the corresponding separate clean signals without noise components as inputs, and the plurality

of adaptive canceling filters receive the corresponding separate noise component signals as inputs.

18. (original) The apparatus of claim 17, wherein the adaptive blocking filters and the adaptive canceling filters are finite impulse response filters.

19. (original) The apparatus of claim 18, wherein the blocking coefficients and the canceling coefficients are updated automatically by an information maximization algorithm.

20. (original) The apparatus of claim 19, wherein a number of taps necessary to implement the feedback structure is optimized.

21. (original) The apparatus of claim 16, wherein the feedback structure of the signal separator comprises:

a plurality of first subtractors that receive the delayed received signals and subtract corresponding signals from the plurality of adaptive blocking filters, and the plurality of first subtractors outputs signals to the plurality of adaptive canceling filters;

an adder that adds signals output from the plurality of adaptive canceling filters to output a total noise component signal; and

a second subtractor that receives the combination received signal and subtracts the total noise component signal to output a clean signal without noise components, wherein the plurality of adaptive blocking filters receive the clean signal without noise components as an input and the adaptive blocking filters generate signals corresponding to a portion of the clean signal without noise components of the delayed received signals to the plurality of first subtractors.

22. (original) The apparatus of claim 21, wherein the adaptive blocking filters and the adaptive canceling filters are finite impulse response filters.

23. (original) The apparatus of claim 22, wherein the blocking coefficients and the canceling coefficients are updated automatically by an information maximization algorithm.

24. (original) The apparatus of claim 23, wherein a number of taps necessary to implement the feedback structure is optimized.

25. (original) A method of removing noise from time delayed signals subject to noise, comprising:

receiving signals having noise components;

delaying the received signals having the noise components by a predetermined period of time to generate delayed received signals;

adding the delayed received signals to generate a combination received signal;

generating separate clean signals without noise components using adaptive feedback filtering based on the delayed received signals, the combination received signal, and the separate clean signals; and

generating a clean signal without noise components using the separate clean signals.

26. (original) The method of claim 25, wherein using adaptive feedback filtering comprises:

generating separate clean signals without noise components by subtracting noise components, output from adaptive canceling filters having predetermined coefficients, from the combination received signal;

generating separate noise signals by subtracting signals output from adaptive blocking filters having predetermined coefficients, which receive the separate clean signals, from the delayed received signals.

27. (original) The method of claim 26, wherein generating the clean signal without noise components comprises adding the separate clean signals.

28. (original) The method of claim 26, further comprising:

updating the coefficients of the adaptive canceling filters and the adaptive blocking filters without signal level information.

29. (original) The method of claim 26, further comprising:

updating the coefficients of the adaptive canceling filters and the adaptive blocking filters automatically by an information maximization algorithm.

30. (original) The method of claim 26, further comprising:

updating the coefficients of the adaptive canceling filters and the adaptive blocking filters automatically by one of a least square algorithm and a normalized least square algorithm.